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ARMSTRONG LABORATORY

A VIRTUAL ACOUSTIC ORIENTATION INSTRUMENT

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Virtual instrumentation developed to create acoustic stimuli for use as orientation signals and the rationale for selecting these target signals is described. A commercially available computer-assisted software engineered programming system, LabVIEW, was used to develop the acoustic signals and the manner of their presentation. Two different acoustic signals are presented. One sound seems to change pitch without limit, and the other is a narrow band-pass noise. The pitch, loudness, and lateral position of the signals are controlled by different functions. Listeners can control the variations in pitch and lateralization and also the rate at which these features change. Programs to generate the signals with sinusoids or with noise bands are described. LabVIEW representations of the programs are included with text descriptions.

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A VIRTUAL ACOUSTIC ORIENTATION INSTRUMENT

INTRODUCTION

The virtual instrumentation described in this report was developed to test the proposition that acoustic representation of data about spatial location can be used to maintain one's orientation in space. In our usual orientational space, determined by the input to the visual, vestibular and kinesthetic sensory systems, the location of an acoustic event is immediately apprehended as right or left (azimuth) and above or below (elevation) one's own position, and, usually, the head rotates in order to fixate the source visually. The distance of an acoustic event is estimated with less precision than estimates of azimuth or elevation. Listeners may select an acoustic event or signal in a background of sound, and locate it, depending upon its salience. For example, speech sounds are prominent when presented at comfortable loudness levels, resist interference by background noise, and listeners may approach them, rather than avoid them as they might avoid sounds with high annoyance. However, speech sounds require linguistic processing and must be understood. Sounds which share the salience of speech, but without the burden of language, should be useful acoustic objects to transmit information about one's spatial location, particularly when there is some uncertainty in its visual assessment.

Although the anatomical and physiological properties of the human and animal systems that mediate the analysis of complex acoustic signals are incompletely described, neural mechanisms that select azimuth angle have been isolated and the alterations in the spectra of acoustic signals by the anatomical features of the external ear have been measured. Neural mechanisms that might support the perception of timbre or quality of acoustic signals have not been described. Indeed, only recently, with increases in computational power, have investigators been able to isolate features of complex acoustic stimuli that neural mechanisms might require for classifying sounds as salient. Information-bearing acoustic signals, such as speech or the echoes detected by hungry bats, are usually complex; i.e., they contain many frequencies with varying amplitudes. The temporal and spectral features of these signals determine their salience and efficiency in transmitting information. The fast computation rates now available for digital signal processing provide the opportunity to synthesize salient and efficient acoustic signals to inform listeners about the space in which they are moving, i.e., in real time.

Recent studies of discrimination among complex acoustic signals have shown that human subjects recognize the patterns of intensity among the frequencies, i.e., the timbre of the signals. The increment in intensity required to detect a change in loudness for a sinusoid, presented alone, cannot be generalized to the detection of intensity increments of the sinusoid presented within a complex of several sinusoids. Indeed, intensity discrimination for the sinusoid is more acute within the complex than when it is presented in isolation (Green, 1988). The intensity pattern among frequencies is an important acoustic feature for synthesizing salient acoustic signals.

For spatial resolution, the range and precision available to the listener is directly related to the spectrum of the acoustic signal. Source location is ambiguous for sinusoids when the interaural difference in time (phase) matches the period of the sine wave, until the frequency becomes high enough that the head provides sufficient sound shadow to produce an interaural intensity difference proportional to azimuth. Broadband acoustic signals allow interaural time differences and intensity differences to be processed simultaneously by the listener, and when the signal bandwidth exceeds 5 kHz, the convolutions within the pinnae, along with reflections from the head and shoulders, impose unique amplitude distributions which are associated with different elevations. Thus, broadband stimuli are localized with greater accuracy than sinusoids in both azimuth and evaluation.

One should be able to synthesize information-bearing acoustic signals that are resistant to masking and common hearing deficits and that are perceptually appropriate and representative of environmental events. The variations in location, pitch, and loudness of the acoustic indicators with spatial information should be immediately apparent. The objective of this project is to provide the pilot with acoustic data that are perceptually consistent with usual indicators for airspeed, vertical velocity, and bank angle. For this application, the acoustic signals must be scaled so that their rates of change are uniquely related to the changes in the airspeed, vertical velocity, and bank angle of an aircraft.

SYNTHESIS OF AUDITORY SPACE

When the sound pressure produced by a broadband acoustic source enters a listener's auditory canals, it has been modified by several features: the space in which the source and listener are located, the azimuth and elevation of the source, the reflective surfaces of the head and shoulders of the listener, and the whorls and curves of each pinna. The pressure waveforms of sounds at each ear for many different source locations can be recovered with small microphones near the tympanic membranes, then digitized and stored in computer memory. These pressure waveforms contain all the acoustic information that was available to the listener when the sound occurred in that space. The Fourier transforms of the digitized voltage waveforms can be used with other spectra, such as microphone characteristics, frequency response curves for loudspeakers, etc., to isolate the spectral properties uniquely associated with source location for a specific individual. The pair of spectra for the left and right ears of a particular subject corresponding to a particular source location is a "location filter." The spectra of new, incoming sounds, corrected for earphone characteristics, can be multiplied by the location filters so that the spectrum for a particular location can be reconstructed, converted to analog voltage, and presented to a listener through earphones. The new, incoming sounds now have the spectral characteristics required for that sound at that location. When the selection of the pairs of spectra, the location filters, is made with respect to head position, one can synthesize auditory localization so that one can "look" at the source location even though the sound is heard through earphones. Ordinarily, sound heard through earphones (as for a conventional stereophonic player) moves as the head moves, but a sound heard "outside" (no earphones) remains stationary when the head moves, and the eyes can turn toward it. If a single pair of spectra were presented, i.e., processed through a single location filter, the source would "move" with the head. In the synthesis of auditory space, head position is usually monitored with a magnetic tracker and the location filters are selected with the coordinates output by the tracker.

The spectra derived from the auditory canal recordings produced by sources at many locations are often termed head-related transfer functions (HRTFs). When the HRTFs are recorded in an anechoic space, they are independent of any particular (echoic) space and different representations of echoic spaces might be added to the HRFTs in order to synthesize sounds heard in concert halls, for example. On the other hand, spectra may be recorded in a particular concert hall and the HRTFs would incorporate the properties of that particular space. The HRTFs must be derived from a broadband acoustic source so that all frequencies that contribute to the perception of location are represented. If the signal to be passed through the location filter has no energy in a frequency region important for a given spatial attribute, e.g., 5-9 kHz for elevation cues, elevation attributed to the sound may be in error. If a listener has poor hearing in the frequency range important for elevation, there may be a loss of resolution. There is still some uncertainty whether an average HRTF has sufficient precision for general use or whether each individual's location filters must be generated in order to synthesize auditory space for that person. Sufficient precision might be available from HRTFs for small, medium, and large pinnae. Imprecise HRTFs produce sound images inside the head, i.e., lateralized rather than externalized as localizations, or their location is misassigned.

The HRTFs specify the original waveforms recorded at the ear canals by calculating the magnitude and phase for each frequency in the broadband signal. A different phase spectrum can be substituted for the calculated one, and another waveform can be recovered by combining the original amplitude distribution with the new phase distribution. For example, the absence of any effect upon the localization of sounds for which phase functions are set to zero for frequencies above 2 kHz would suggest that the listener does not use time differences at high frequencies. Kistler and Wightman (1992) analyzed HRTFs into their principal components and found that five components are required to synthesize functions so that subjects can perform at 90% correct assignment of location for synthesized sound sources. For frequencies below about 4 kHz, the amplitudes of the five functions are all quite similar. For frequencies above 5 kHz, the amplitudes are substantially different. The amplitude of component 1 decreases very slightly with frequency from 200 Hz to 15 kHz. When only component 1 was presented, azimuth angle was well discriminated, but the subjects reported a high number of front-back confusions. Component 1 accounted for 70% or more of the variance for each of the 10 subjects. There are substantial variations in amplitudes at the higher frequencies for components 2-5. As higher order components were added, the number of confusions reported by subjects decreased and elevation was more accurately detected.

In a companion paper, Wightman and Kistler (1992) were able to control interaural time difference (ITD) for the spectra delivered to earphones. Thus, a time cue appropriate for one location could be added to a spectral shape appropriate for another. Presumably, the dominant cue would determine the perception of location. Subjects' assignments of location conformed to the direction corresponding to the ITD. In a second experiment, the stimuli were high-pass filtered at 5, 2.5, 1, 0.5, and 0.2 kHz. For the 5-kHz, high-pass condition, the judgments followed the direction indicated by the spectral shape and the interaural intensity differences. Thus, as the cut-off frequency decreased, the ITD cue became more prominent. Wightman and Kistler (1992) suggest that, for wide-band stimuli, the ITD determines the range of possible locations and that the spectral shape and interaural intensity difference cues then disambiguate that range to specify source location accurately. Thus, the entire spectrum is used by the auditory system.

DESIGN OF ACOUSTIC ICONS

The broad frequency representation required to simulate auditory space fits well with the requirements for generating distinctive acoustic signals, or icons, that can provide information about environmental events. The broad frequency representation insures that all individuals will be able to perceive the icons; however, a question remains about the perceptions of listeners with reduced sensitivity at some frequencies. Individuals with hearing losses in frequency regions that mediate information about, say, elevation, may not perceive synthesized elevations in the same way as do others without such losses. In actual situations, cues are often redundant and the loss of one acoustic cue among many in a particular situation may not be critical.

For the present application, two acoustic objects have been constructed. One of them is a narrow band noise, 0.4 octave wide; its center frequency varies from 500 to 4500 Hz. The pitch of the sound changes as the center frequency increases and decreases. The second acoustic object is constructed from a recipe given by Shepard (1964) for a sound that appears to increase (or decrease) continuously in pitch, yet does not seem to arrive at a destination. The signal is constructed from a series of 10 sine waves, at 1-octave increments; the amplitudes of the middle components are greater than those at either end. Twelve sets of 10 sine waves each are generated. The frequencies in each successive set are incremented by the 12th root of 2 so that the highest frequency of the twelfth set is one octave above the highest frequency of the first set. The twelve sets are output successively, and the series begins again with the first set after the twelfth set is output. The pitch of the complex sound continues to increase (or decrease) as the sets of twelve sine waves are cycled, even though the sets are repeated. Taken together, the twelve sets constitute a pitch circle. Although the pitch ranges for the Shepard illusion and the narrow-band noise are similar, the two acoustic objects are distinctly different in quality and cannot be confused even though both sounds are presented simultaneously. The pitches of the two sounds may increase together or move in opposite directions. The rate of change in the pitch for each sound can be independently controlled. When the Shepard illusion is built from narrow bands of noise (0.2 octave wide) rather than sinusoids, the two acoustic objects are more similar in timbre, but are still clearly distinguishable.

When the listener has earphones available, interaural intensity differences can be superimposed on the narrow band noise so that the lateralization of the image produced by the binaural stimulation can be varied. In this condition, the sound image produced by the noise moves from one side of the head to the other, i.e., lateralizes, and the image produced by the Shepard signal remains at the center of the listener's head. The listener can click on a switch and apply the interaural intensity difference to the Shepard signal also, and both sounds move together from one side of the head to the other. The perceptual separation of the two objects is clearer when the interaural intensity differences are applied to one sound, rather than to both, particularly when the Shepard illusion is constructed from the 0.2-octave noise bands.

Acoustic signals similar to these have been used in our earlier studies as aids to orientation in a flight simulator and in flight tests. In those tests, lateralization was associated with the bank angle of the aircraft (Lyons, et al., 1990), the pitch of the band-pass noise with airspeed and the rate of pitch change of the Shepard illusion with vertical velocity (Gillingham and Teas, 1992). The present software attempts to maximize the salience of the orientation signals.

ACOUSTIC ORIENTATION SIMULATOR PROGRAM

Shepard (1964) created the pitch illusion using a special program written for an IBM 7094 computer. The 7094 output was recorded on digital tape that was then played back, off-line, to make an audio tape that could be played at 7-1/2 in./sec on an audio tape player. For the present application, we have used a computer-aided software engineering program, LabVIEW, residing on a Macintosh llfx personal computer equipped with a digital signal processor, to generate programs that calculate the acoustic signals and then output them to earphones. These programs are included in the Appendixes. Since the acoustic output of the Macintosh is generated on-line, it can be varied as the program responds to events read by analog/digital converters, or to pulses from external equipment in real time. At present the controls for the acoustic output are located on the Front Panel of the program. In the final realization of the acoustic orientation instrument, the acoustic signals will be controlled by the digitized representations of flight parameters, e.g., airspeed, vertical velocity, and bank angle. (Note: For the flight tests run previously with earlier versions of the programs, the analog control signals were digitized before delivery to the Macintosh. With the present configuration of the Macintosh, the analog signals representing flight parameters can be taken directly to the computer for digitizing, thus eliminating a time-consuming step in the link between the control signal and the acoustic response to it.)

The Acoustic Orientation Simulator (AOS) program is included in Appendix A. The user designates a prerecorded pitch circle (PC) for retrieval, and selects control patterns for the band-pass noise (BPN) and lateralization (LAT) with the **RangeControlFunctions.** At present the user also controls the timing of the changes in the PC, BPN, and LAT (**TIMING CONTROLS/PULSE GEN'S),** the direction of the pitch change for the PC (**Directn**), the selection to lateralize the BPN only (**Latrlz**, right, and **Latrlz Both**, down) or both the PC and BPN (**Latrlz Both**, up) or to have no lateralization at all (**Latrlz**, left); i.e., both PC and BPN images superimposed in the center of the listener's head. The listener can choose to have the LAT passed through a Flat-Top Window (**Window**) so that the lateralized BPN is pulsed. The listener controls the overall loudness for the PC (**Gain/Ill P**) and the BPN (**Gain/BPN**) independently. In a run time-only version of the AOS, in which flight parameters would be controlling the timing, many of these listener controls and their indicators would be eliminated. The controls are retained in the simulation program so that the effects of variation in the parameters can be studied.

Program Sequence

The Front Panel (Fig. A-1, p. 15) shows the user controls and their indicators. When the user clicks the mouse on the LabVIEW run indicator, a dialogue box appears. The program is now at Frame 0 of the diagram (Fig. A-2, p. 16) and has paused to accept the PC file name and location. The user enters the folder and file name and the program retrieves the PC file, emitting a soft 30 msec-tone at 440 Hz if the retrieval is successful. If it is not successful, a louder, high-frequency tone (2 kHz) is emitted, indicating an error and the program stops. If the retrieval is successful, the program clears, configures, and starts the timers (inner frames 0, 1, and in Figure A-3, inner frames 2 and 3, p. 18).

In Frame 1 (Fig. A-4, p. 19, top), the circular buffer for continuous output of the acoustic signals is prepared with Setup Dduf (See Sub-VIs, Appendix C, for diagram). The first step in constructing the band-pass noise (BPN) is also carried out. Note that the channel designation for Setup Dbuf is "2" which indicates that two channels of the analog-digital converter are to be used simultaneously for binaural stimulation. In this case, the digital representations of the waveforms for each channel are interleaved, i.e., the left ear might receive the values in locations 0, 2, 4, etc., of the buffer, and the right ear, those values in locations 1, 3, 5, etc. Thus, the total number of points for the buffer is the sum of the points for each of the two channels. For generating the low-pass noise, the number of points corresponds to that required for one channel. Due to the interleaving architecture that is built into the LabVIEW software, any operations on one of the two channels must be carried out after separating the interleaved representations. LabVIEW has a sub-VI (an elemental virtual instrument) for "decimating" the interleaved buffer.

After preparing the buffers to accept the inputs, the sub-VI, START, is called in Frame 2 (Fig. A-4, bottom). The functions to control the indexing of the arrays for BPN, PC, and LAT are obtained in Frame 2 from the sub-VI, Pattern Gen, shown in Appendix C (Fig. C-2, p. 36, Fig. C-3, p. 37). The display of control functions on the Front Panel of the AOS is generated in this frame. The Pattern Gen sub-VI outputs a ramp (RMP), a triangular function (TRI), a sine wave (S), and the sum of 2 sine waves (2S) or the sum of 3 sine waves (3S). The RMP must always be used for the PC, but the other two control functions may be selected by the listener. The defaults are designated as 0 (RMP) for the PC, 1 (TRI) for the BPN, and 3 (3S) for the LAT. On the Front Panel of the Pattern Gen sub-VI, a choice can be made to show either the triangular function or the sine function.

Frame 3 (Fig. A-5, p. 20) contains the While Loop within which the operations of timing, indexing, lateralizing, adjusting gain, etc., all take place. Within the While Loop are inner-frame 0 and inner-frame 1 (Fig. A-6, p. 21). On the left and right vertical borders of the While Loop, the small triangles indicate shift registers. On the left, the shift registers (apex down) are initialized to zero. The shift registers on the right (apex up) save the incremented indices, and pass them back to the shift registers on the left. The shift registers track the indexing of the arrays of the PC, the BPN, and the LAT. At the top of inner-frame 0, the sub-VI, DECl, receives the interleaved array from the PC file and separates it into two output arrays, one for the left and one for the right ear. When the ascend/descend switch is True, the indexing array, brought from Frame 2 into Frame 3, is stepped in ascending order to produce an increasing pitch of the PC. When the switch is False, the array is reversed, and produces a decreasing pitch. During this operation, the counter reads the pulse generator output and when that count matches the control count entered from the front panel for PC, the large Boolean to the right of the figure is selected True, and the counter is started again. The index (from the shift register on the left) is incremented (+1) and the value is checked for validity; i.e., it should be less than the size of the array. If the index is valid, the small, inner Boolean reads True and the index is passed to the shift register on the right, to the index array function (to the left of the inner frame 0) and to the index input on DECl. The two PC arrays, separated into Left and Right, are passed out of inner frame 0 to inner frame 1. If the count is not reached and the large Boolean on the right reads False, no change in index occurs. If the index exceeds the size of the index array, the small Boolean is False and an index of 0 is set; i.e., the index array is started again at its beginning.

The program flow for BPN is similar except that the index array is used somewhat differently. The amplitude of the control function is used to determine the center frequency of the BPN. The sub-VI, **CNTROL**, establishes the range of center frequencies (500-4500 Hz) and converts the amplitude of the control function at the given index number to a frequency. The sub-VI, **Cmpnts Varifreqy** BPN, receives the frequency and calculates upper and lower cut-off frequencies for the bandpass filter, and establishes the gain. The BPN is then taken to the right edge of the inner frame 0 to be passed to inner frame 1.

The timing function for LAT is carried out in inner frame 1. As was the case for BPN, the indexing for LAT is based on the magnitude of the control function. The amplitude of each value in the waveform on the Front Panel, e.g., the sum of 3 sine functions, is delivered to the sub-VI, IID ARRAYS, and may be applied to either the BPN, the BPN and the PC, or to neither, according to the selection. As in inner frame 0, if the count has not reached the control count, the large Boolean is False and the signal passes through. If the count is reached, the case is True and the index figure is checked for validity. If the Latrlz switch is closed, then the value is passed to the index input of the IID ARRAYS, and the BPN, whether it is passed through the Flat-Top Window or not, is input to the sub-VI, along with the two channels of the PC. Within the sub-VI, IID ARRAYS, the selection to lateralize both PC and BPN or only the BPN is made and only Left and Right channels are output. The two channels are then interleaved and taken to BLK LOAD for output through the digital-analog converter to the earphones.

When the **STOP** switch on the Front Panel is momentarily closed, the While Loop terminates and the program moves to Frame 4, which contains the sub-VI, CLEAR. The circular buffers that hold the digital representations of the most recent orientation signals are cleared and the program stops, having completed the final frame.

GENERATION OF PITCH CIRCLES

The AOS program reads in a file for the pitch circle (PC). Two programs, Illusory Pitch/Sines and Illusory Pitch/Noise, may be used to generate the file that is read by AOS. These programs, included in Appendix B, share many features with the AOS, but are richer in displays, and, in addition to the acoustic output, show the waveforms and their spectral properties in graphic form. Pitch circles can be generated for different frequency regions, with different numbers of components, with different frequency increments, etc. Different amplitude distributions might also be explored by altering the equation used to calculate amplitudes.

Illusory Pitch/Sines

Figure B-1, p. 24, shows the Front Panel for the program, Illusory Pitch/Sines. Several parameters are specified by the controls shown on the Front Panel. **F-min** (10) establishes the low frequency limit (Hz) of the sines, and **C-max** establishes the number of frequencies (10). **T-max** (12) controls the number of incremental steps. **L-min** (12) and **L-max** (56) establish the voltage range in decibels for the magnitudes of each sine wave (each C). These values are used in Frame 1 to calculate the frequencies and amplitudes of the sine waves used to construct the PC.

There are also controls for establishing the **Time Base** (24,000), the **T Base Code** (4), the **No. Pts** (8192), and a delay, **Ticks wait** (60). At the upper right of the Front Panel there are gain controls, **Common Gain** (1.00) and **Multiplier** (1), and the PC array, **PCrcl Array** to read out individual elements. In addition, several error indicators were included to provide information as the program was being developed. There is also provision for changing the time base while the program is running (**Timebase chnge**).

The upper of the two small graph indicators to the right shows the voltage waveform for each collection of 10 sine waves, i.e., each t of **T-max**. The lower one shows the intensities for each sine wave in the collection of 10 sine waves. The displays follow the computation of each t. When all the calculations are completed, there are 12 arrays of 10 sines each. The voltage waveform for each of the 12 is displayed as it occurs in the upper of the two large graph displays on the left of the Front Panel. If the **PWR Sp?** switch is pressed, the power spectrum of the voltage waveform for each C is shown in the lower graph indicator on the left. The toggle switch at the lower left (**Up=ascend**) controls the direction of movement through the C arrays.

The controls and indicators on the Front Panel are used in the diagrams representing the program. Frames 0 and 1 are shown in Figure B-2. In Frame 0, the beginning of the program, a sine pattern is used to set up the buffer for continuous output. This operation was also described previously for the AOS. In Frame 2 the elements of the PC are calculated previously using For Loops and Formula Nodes as structures within which the calculations are carried out. The values entered into the controls on the Front Panel, i.e., **T-max, F-min**, etc., appear at the left, within Frame 1. There is an outer For Loop and an inner For Loop. The number placed in the N in the upper left corner of each Loop determines the number of iterations. The value for **T-max** is wired to the N of the outer For Loop and the value for **C-max** is wired to N of the inner For Loop. Each time the outer loop executes (each t), the inner loop executes C-max (10) times. F-min is taken to the left border of the Formula Node within the inner For Loop, and **T-max** is also wired to the Formula Node. Each time the For Loop executes, the i in the lower left corner is incremented and is used to represent the current value of t. It, too, is taken to the Formula Node. These values are used by the formula expressed within the Node. The output from the calculations can be accessed on the right edge of the Formula Node structure. In a similar way, quantities are taken to the left edge of the lower Formula Node to calculate the amplitudes of the sine waves which will be collected in an array, representing that set of Cs, i.e., each t.

The calculated frequency is taken to the <u>cycles</u> input of the sub-VI, Sine Pattern, and the amplitude is taken to the <u>amplitude</u> input. The number of points has been imported from Frame 0. The calculated values are summed into the Sine pattern and delivered to a shift register on the right edge of the For Loop and displayed on the Front Panel. The next iteration of the outer For Loop can then begin and the set of 10 Cs for the next t is calculated and added to the previous calculations.

When T-max is reached, the program moves to Frame 2 (Fig. B-3, p. 26) and the instruction, START, is given the circular buffers. In Frame 3, the switch, **Save Pitch Circle,** is checked, and, if True, the dialogue for saving the PC is carried out and the file is saved (Fig. B-4, p. 27). If the switch is False, no file is saved and the program moves on to Frame 4.

Frame 4 (Fig. B-5, p. 28) contains the While Loop which allows the program to run continuously until the **STOP** switch on the Front Panel is toggled. Within the While Loop there is a For Loop, and **T-max** is wired to its N. At the left, within the For Loop, the **Up=Ascend** switch controlling the direction of the PC is checked. If True, the array is addressed in ascending order, and each i increments the array index. If the switch is False, the array index is decremented. The sub-array that is selected by the index is interleaved at the output of the Index-Array sub-VI. The interleaved array is separated into two channels with the Decimate sub-VI, and one channel is displayed and also taken to the Power Spectrum VI. If the **PWR Sp?** switch is True, the power spectrum is displayed on the Front Panel. If False, no display occurs. The program provides the opportunity for the values in the other channel to be multiplied in order to create an interaural intensity difference which would move the image to one side of the head. The two channels are then interleaved again and another multiplier is available to increase the voltage of both channels, i.e., to make both channels louder. If the direction has not been reversed, the Boolean is False and the interleaved arrays are taken to the BLK LOAD sub-VI which outputs the waveform from the digital-analog converter and the listener can hear the sounds. In the True case, the direction has been reversed and the previous output is continued. Within the For Loop the RATE sub-VI is checked and, if it is different, the new rate becomes effective. The **STOP** switch on the Front Panel is also checked and, if True, the program exits from the While Loop and moves to Frame 5.

In Frame 5 (Fig. B-6, p. 29), after the sub-VI, CLEAR, is executed, the program stops.

Illusory Pitch/Noise

The program for Illusory Pitch/Noise is similar to that for Illusory Pitch/Sines except for the use of the noise rather than sine waves to synthesize the PC. Figure B-7, p. 30, shows the Front Panel for Illusory Pitch/Noise. The broad peaks in the power spectrum of the noise bands in Figure B-7 contrast strongly with the narrow peaks of the power spectrum in Figure B-1 for Illusory Pitch/Sines. For the noise bands, **F-min** is set at 10 Hz, rather than at 5 Hz as is the sines version. A choice of passing the noise bands through a window to reduce the onset and offset transients for each sub-array has also been included.

Frames 0 and 1, shown in Figure B-8, p. 31, are also different due to the generation of noise bands. In Frame 0, Gaussian Noise is generated and led to a 10-kHz, 2nd order, low-pass filter. The output of the low-pass filter is taken to Frame 1 and passed through a band-pass filter with high and low cut-off frequencies determined by values based on the frequencies calculated in the Formula Node structure. These frequency limits are derived by multiplying the node output by 1.10 to set the high frequency limit, and by 0.90 to set the low frequency limit. Thus, the bandpass filter is 0.2 octave wide. If the center frequency is 100 Hz, the bandwidth includes frequencies from 90 to 110 Hz, and when the center frequency is 1000 Hz, the limits are 900 to 1100 Hz. For Hz, width is proportional to the center frequency.

In Frame 4 (Fig. B-9, p. 32) the optional window for the noise bands is inserted after the interleaved arrays have been separated. A Kaiser window is used. In other respects, the program to generate PCs with narrow bands of noise is the same as that for sinusoids.

REPRESENTATION OF FLIGHT PARAMETERS

The Pitch Circle was used to indicate aircraft vertical velocity in a flight test (Gillingham and Teas, 1992). Although the strategy for controlling the rate of change in the PC implemented in the current program is different from that used previously, the same relation has been retained. When altitude is increasing, the pitch of the PC increases; when altitude is decreasing, the illusory pitch decreases. When the magnitude of the vertical velocity is increasing, the rate of change in illusory pitch increases; and when the magnitude of the vertical velocity decreases, the rate of change in illusory pitch decreases. In the flight tests, the PC was absent when vertical velocity remained constant within +/-100 ft/min. When the vertical velocity threshold values were exceeded, the rate of pitch change varied with vertical velocity. In the current program, the PC is heard continuously and the listener controls the rate and direction of the PC on the Front Panel. In subsequent tests of the present scheme -- in a flight simulator, for example -- the voltage representing altitude would be read continuously. When the change in voltage within a predetermined time interval exceeded a criterion value, the sign and magnitude of the difference would be used to control the direction and rate of change, respectively, of the voltage output representing the PC. By substituting a pattern for the voltage readings from the simulator, one could implement, at a computer console, a sequence of changes similar to those generated by the flight simulator.

The band-pass noise (BPN) was used in the flight tests to indicate airspeed. As airspeed increased, the center frequency of the noise also increased. In the present realization, 15 steps from 500 to 4500 Hz are available. These limits and step sizes can be altered. Since the BPN is always audible, its lateralization was used to indicate bank angle. In the flight simulator, the voltage representing bank angle would be read continuously and used to control the intensity at each ear. An important consideration is the relation between bank angle and the amount of interaural intensity difference. An interaural intensity difference of 12-15 dB fully lateralizes a sound. Lyons et al., (1990) using a different acoustic signal from that described here, found that pilots preferred the full interaural intensity difference to be displayed for +/-3 degrees of bank angle. The detection of a small deviation in bank angle was highly desirable in that particular experimental setting, in which the desired performance was straight and level flight. If the full dynamic range of lateralization is used for a small deviation, the discrimination of larger bank angles is not available. The mapping of bank angle to lateralization should be optimized according to type of aircraft and mission.

Even though each of the flight parameters may be made to change independently, the more interesting situation is for simultaneous variation along the three auditory dimensions. The present simulation program presents such simultaneous changes so that one can perceptually track the variations in each dimension. However, there is as yet no feedback control to offset programmed changes in the acoustic dimensions, as there might be for maintaining a steady state in a flight simulator. By incorporating a joystick control, one should be able to present acoustic patterns in which the three acoustic dimensions vary simultaneously and to ask subjects to maintain one, two, or three of the dimensions in a steady state. By tracking the mean and variance of the variables controlling the actual acoustic output, the performance of the subjects could be measured for different characteristics of the acoustic parameters, in order to determine the limits within which the acoustic parameters offer useful information. Once the limits of performance are established, the effects of distraction upon the performance of the subjects could also be measured.

REFERENCES

1. Gillingham, K.K., & Teas, D.C. Flight evaluation of an acoustic orientation instrument (AOI). Presented at the Aerospace Medical Association Annual Scientific Meeting, New Orleans, LA, May 10-14, 1992.

2. Green, D.M. Profile Analysis: Auditory intensity discrimination. New York: Oxford University Press, Inc., 1988.

3. Kistler, D.J., & Wightman, F.L. A model of head-related transfer functions based on principal components analysis and minimum-phase reconstruction. J Acoust Soc Am 91:1637-1647 (1992).

4. Lyons, T.J., Gillingham, K.K., Teas, D.C., Ercoline, W.R., & Oakley, C. The effects of acoustic orientation cues on instrument flight performance in a flight simulator. Aviat Space Environ Med 61:699-706 (1990).

5. Shepard, R.N. Circularity in judgments of relative pitch. J Acoust Soc Am 36:2346-2353 (1964).

6. Wightman, F.L., & Kistler, D.J. The dominant role of low-frequency interaural time differences in sound localization. J Acoust Soc Am 91:1648-1661 (1992).

APPENDIX A

LabVIEW Representation of AOS

Acoustic Orientation Simulator
Tuesday, November 3, 1992 12:21 PM

Page 1

Front Panel

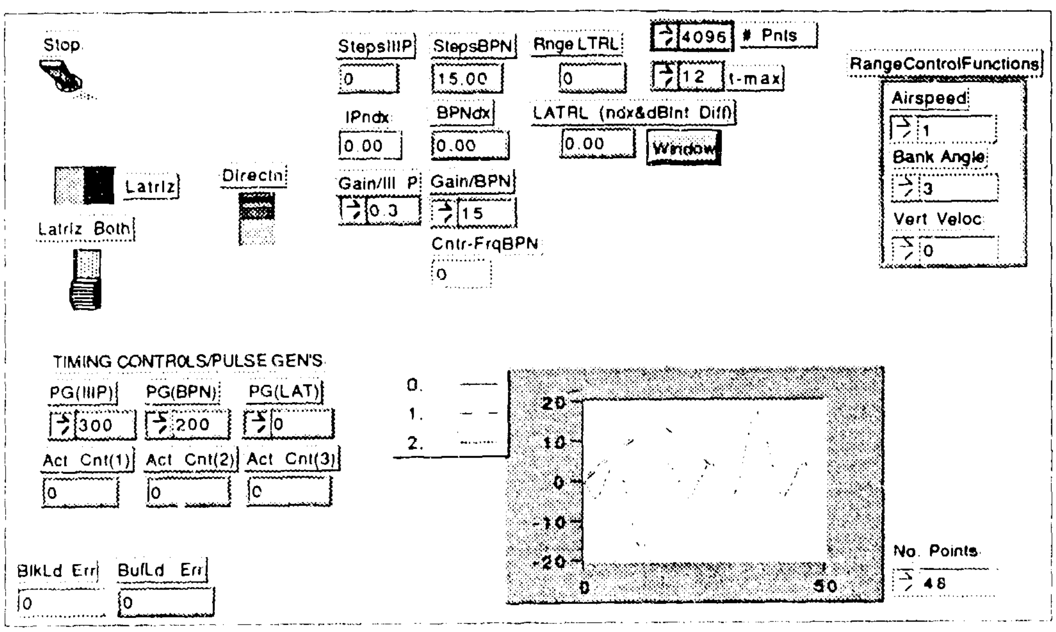


Figure A-1. Front Panel of Acoustic Orientation Simulator (AOS). Description in Text.

Block Diagram

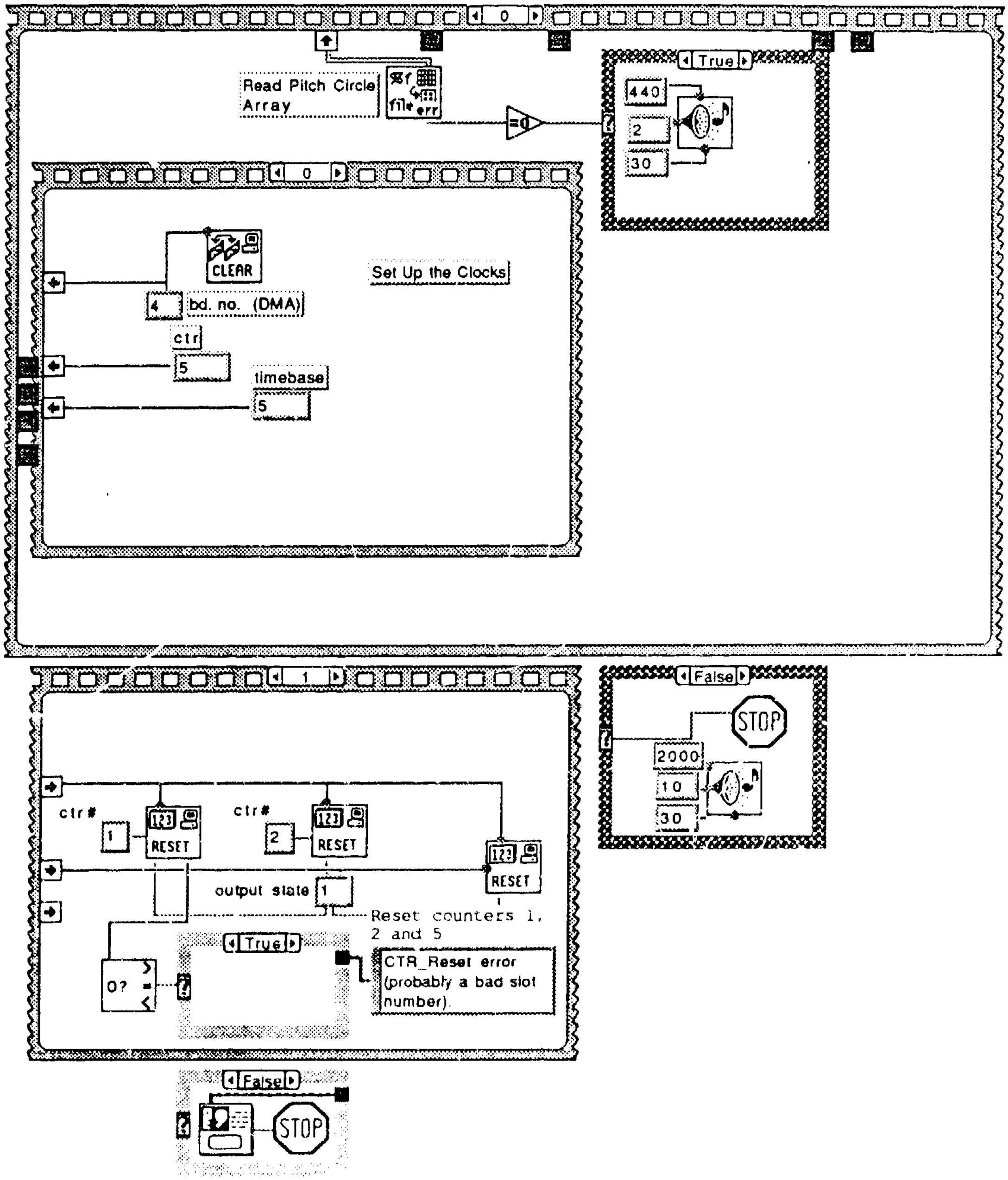


Figure A-2. Frame 0 of AOS. Read Pitch Circle Array, initiate counter routine: clear and reset counters. See text for further description.

Acoustic Orientation Simulator
Tuesday, November 3, 1992 12:21 PM

Page 3

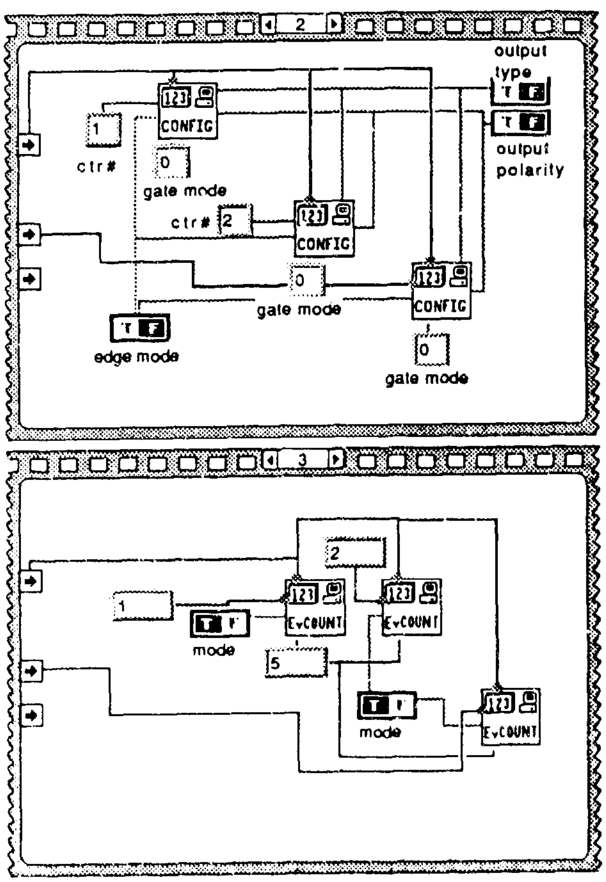


Figure A-3. Frames 2 and 3 within Counter routine: counters are configured and counting is started.

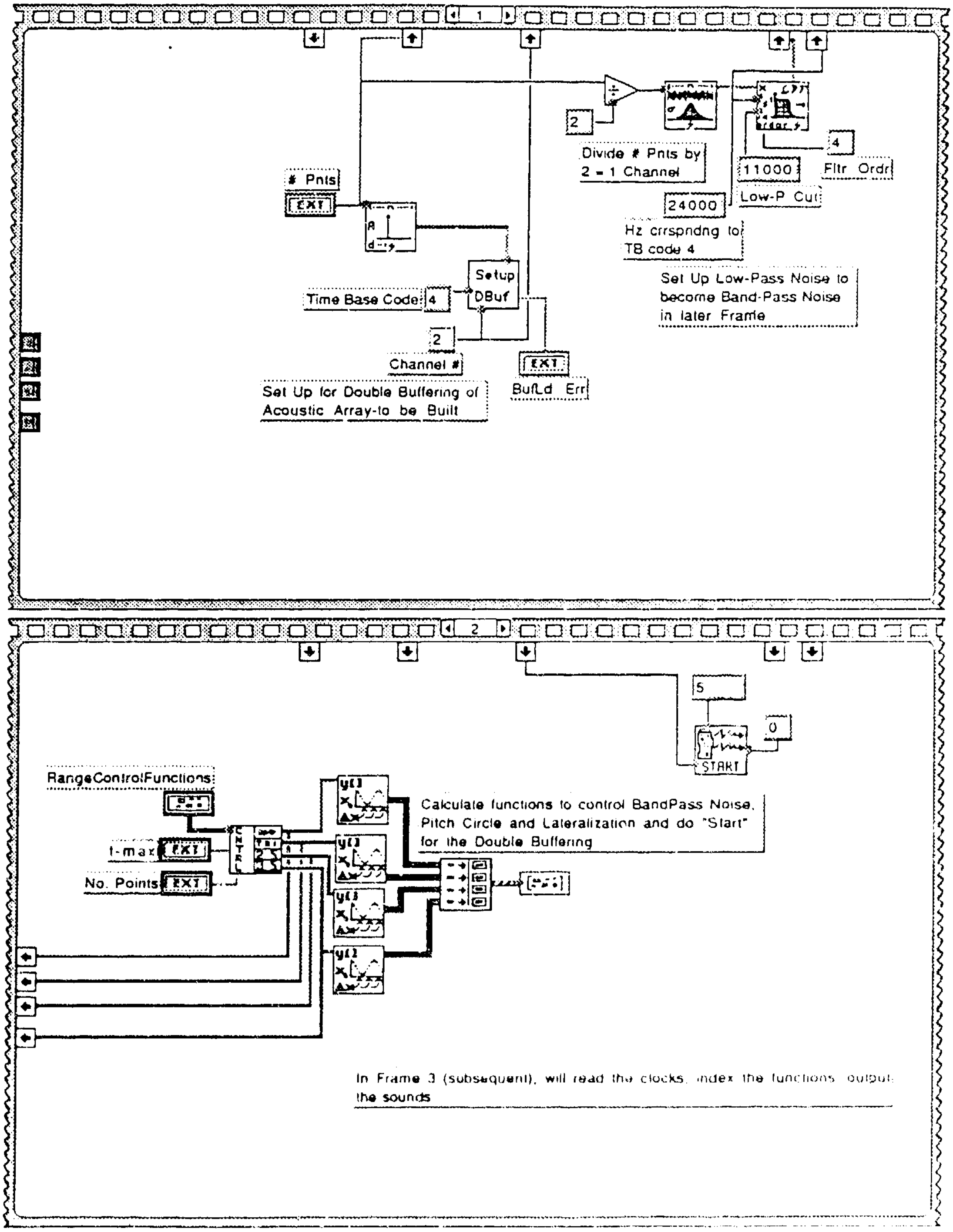


Figure A-4. Frames 1 and 2 of AOS. Frame 1: set up for double buffering of output and generate the low-pass noise from which band pass noise will be formed. Further discussion in text. Frame 2: call Pattern Gen sub-VI, display waveform patterns on Front Panel of AOS.

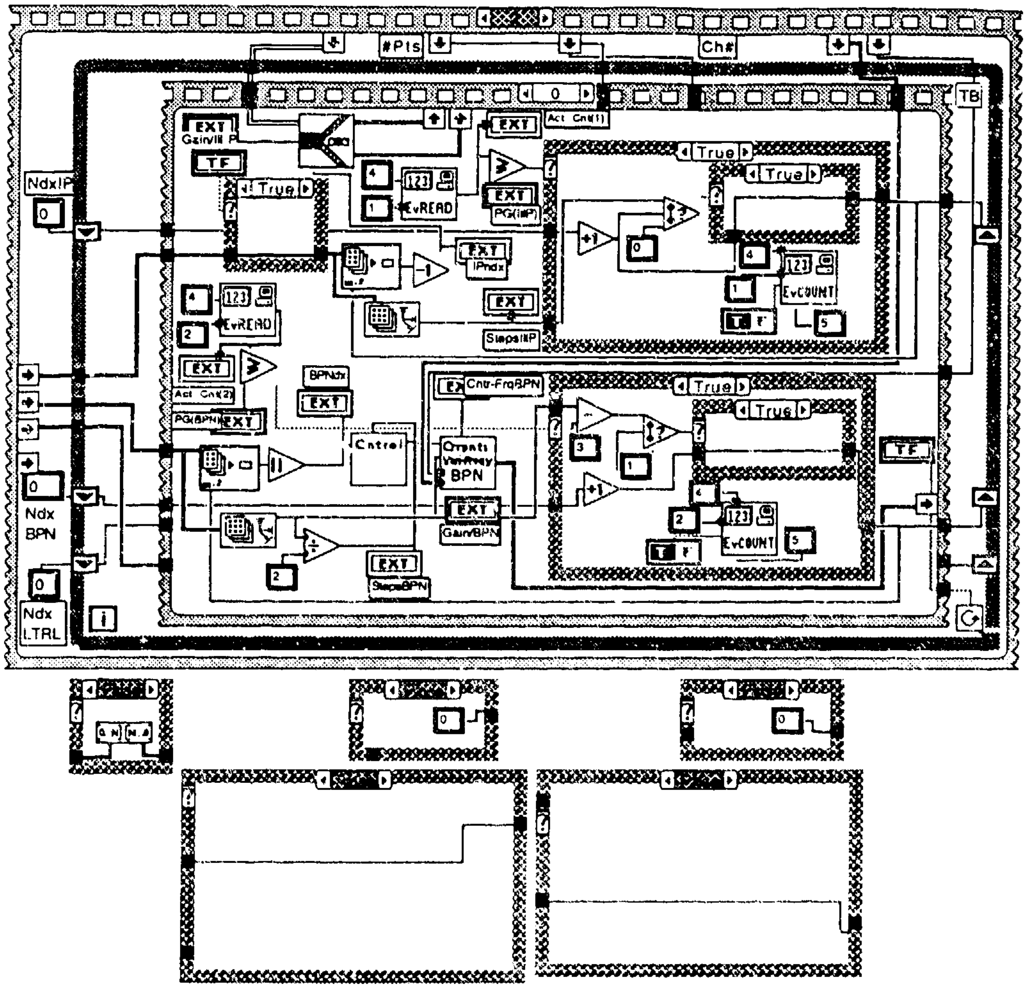
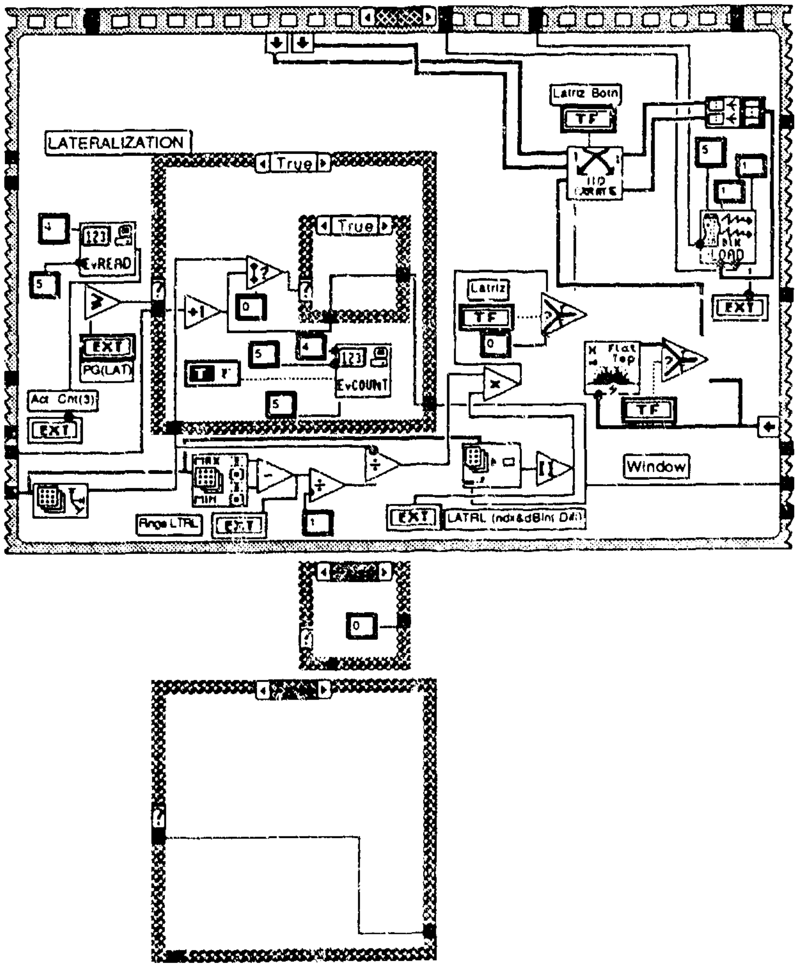


Figure A-5. Frame 3 of AOS. Within Frame 3 is the While Loop that allows continuous operations of the functions contained within the inner Frames 0 and 1. The functions within Inner Frame 0 control the counters, etc., for the PC and BPN. The operation of these functions is described in the text.

Acoustic Orientation Simulator
Wednesday, October 7, 1992 4:10 PM

Page 6



Frame A-6. Inner Frame 1 (within the While Loop). The timing and pattern functions for lateralization of the signals are contained in this frame. In additional, the signals are output to the earphones from the sub-VI BLK LOAD.

Acoustic Orientation Simulator
Tuesday, November 3, 1992 12:21 PM

Page 7

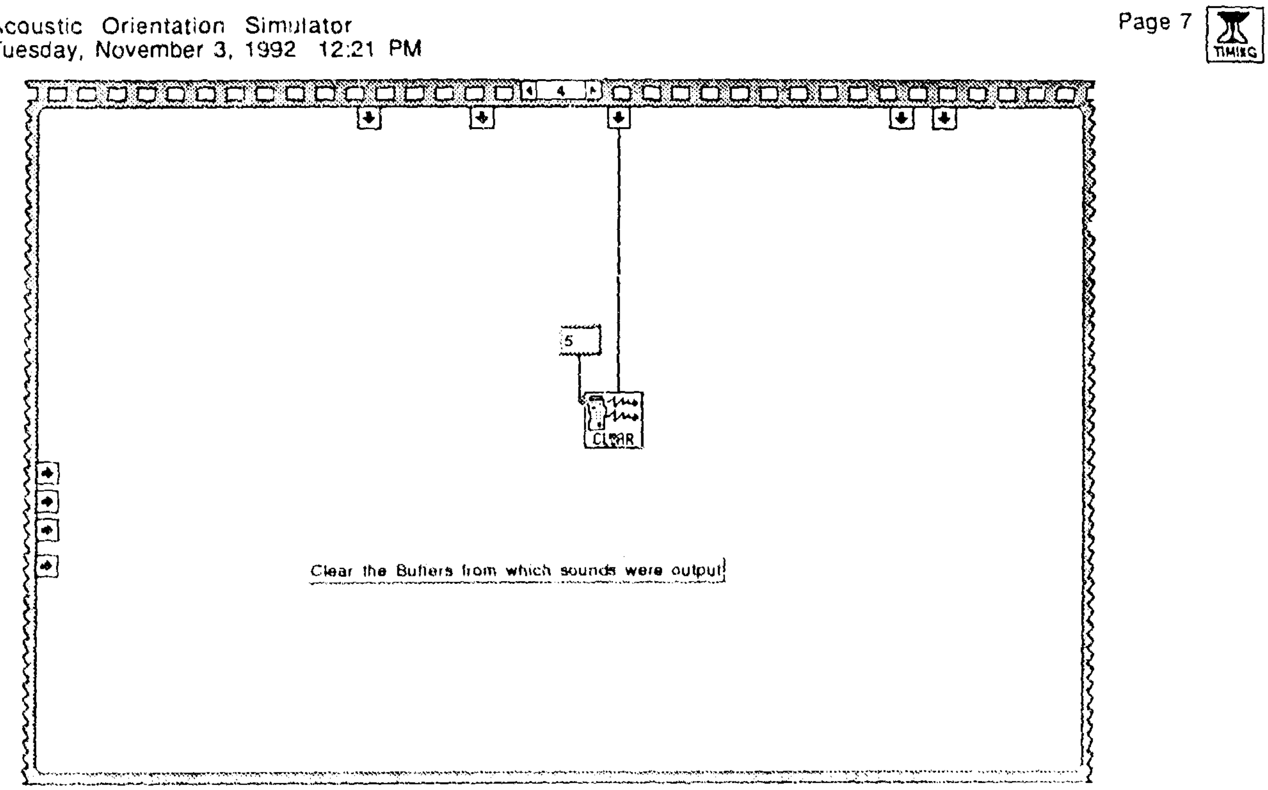


Figure A-7. Frame 4 of AOS. The buffers containing the values for the acoustic signals are cleared. Upon completion of this frame, the program stops.

APPENDIX B

LabVIEW Representation of Illusory Pitch/Sines

LabVIEW Representation of Illusory Pitch/Noise

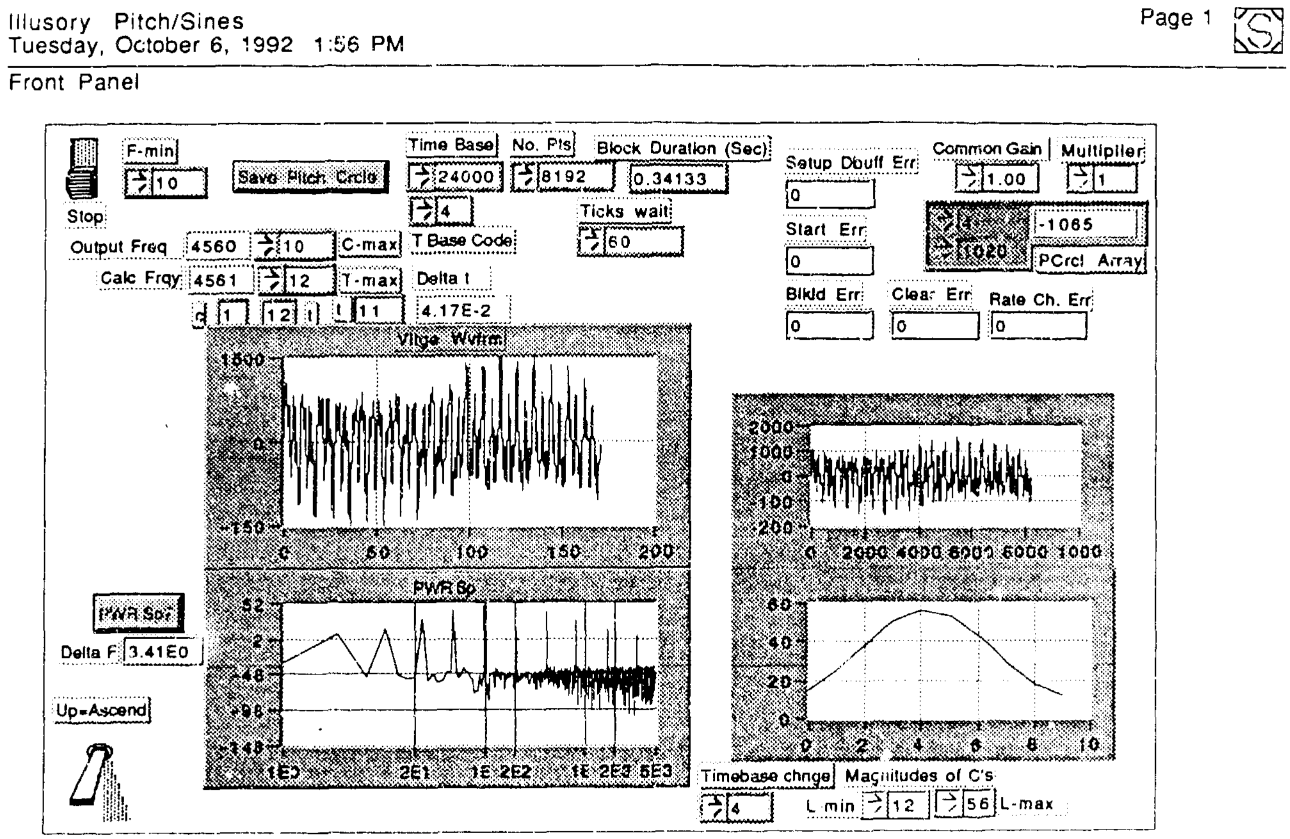


Figure B-1. Front Panel of the program for generating the Pitch Circle with sine waves. Controls and displays are discussed in text.

Block Diagram

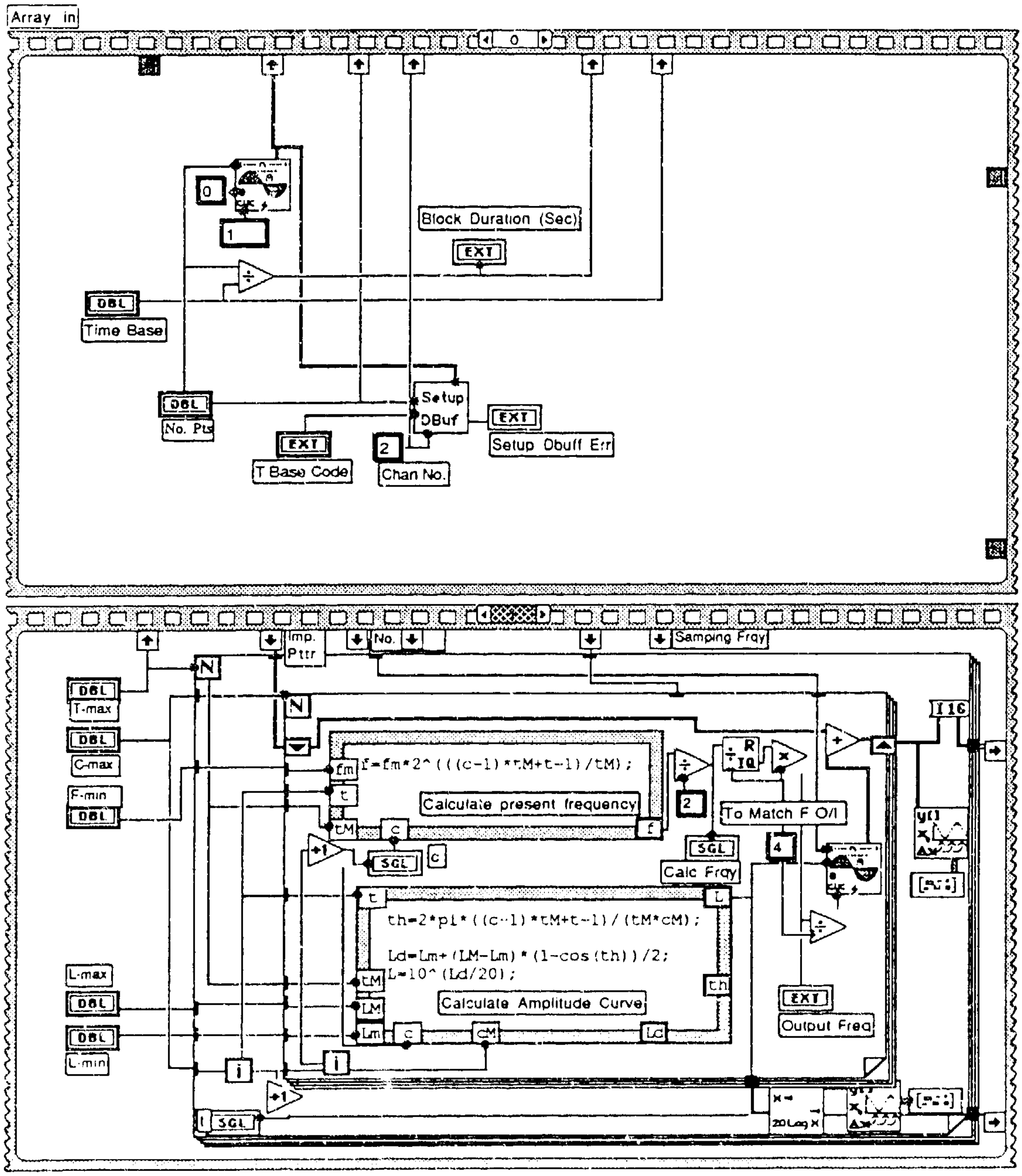


Figure B-2. Frames 0 and 1 for Illusory Pitch/Sines. In Frame 0 the buffer is set up for continuous output of the PC. In Frame 1, the frequencies and their amplitudes are calculated and the displays of the voltage waveform for each t and the amplitudes of each frequency within the array are programmed.

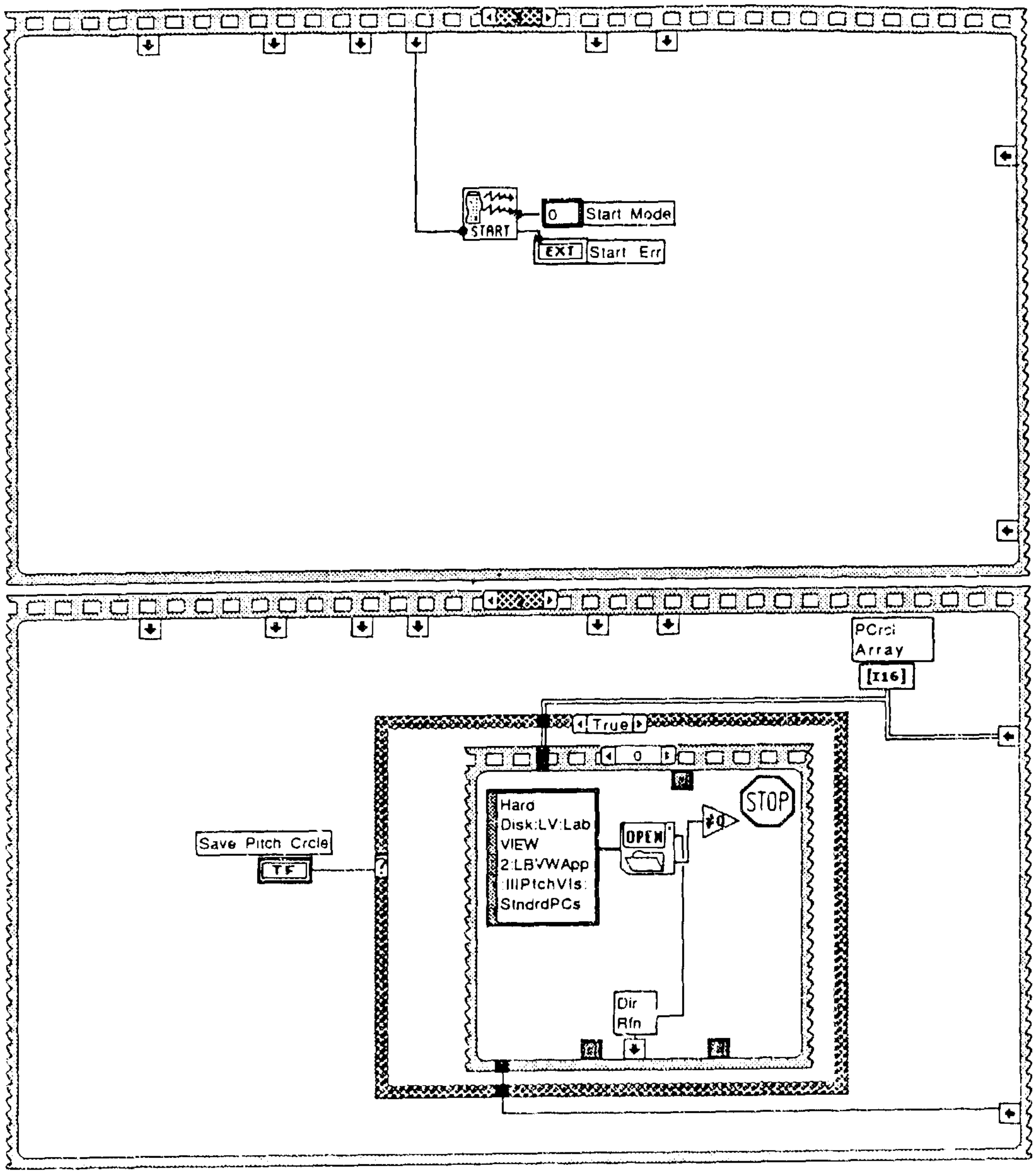


Figure B-3. Frames 2 and 3 for Illusory Pitch/Sines. In Frame 2 the START sub-VI initiates the double buffering. In Frame 3, the switch to store the PC is checked and, if positive, initiates the disk routine to save the digital representation of the PC, to be carried out by the sequence of small frames within the Boolean structure.

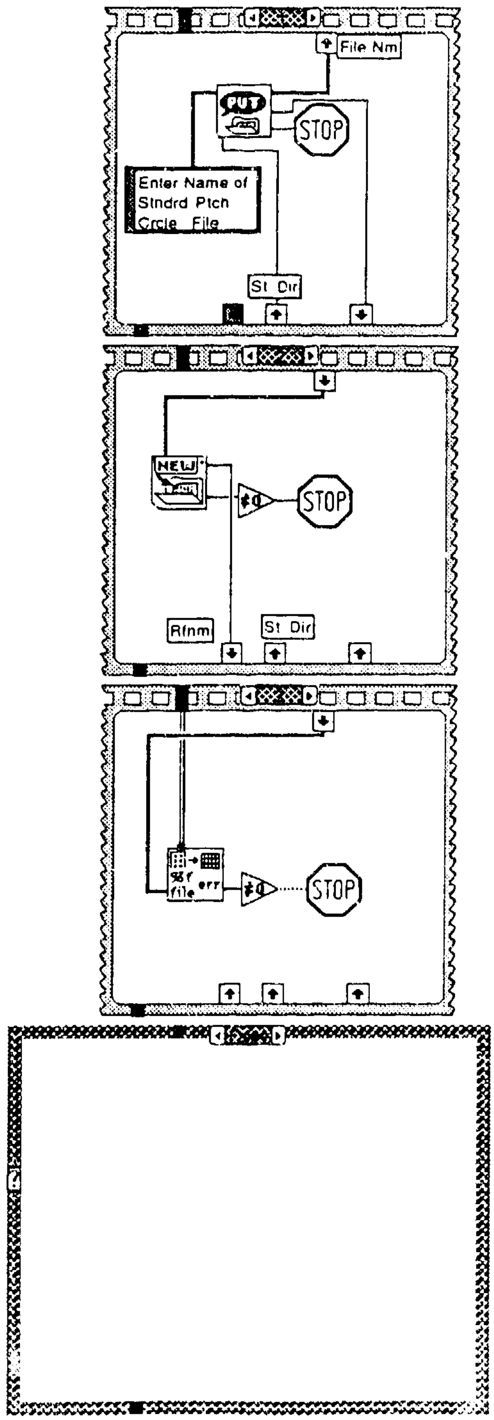


Figure B-4. Completion of the routine to store the PC and the False case for the Boolean structure.

Illusory Pitch/Sines
Thursday, August 13, 1992 2:43 PM

Page 5

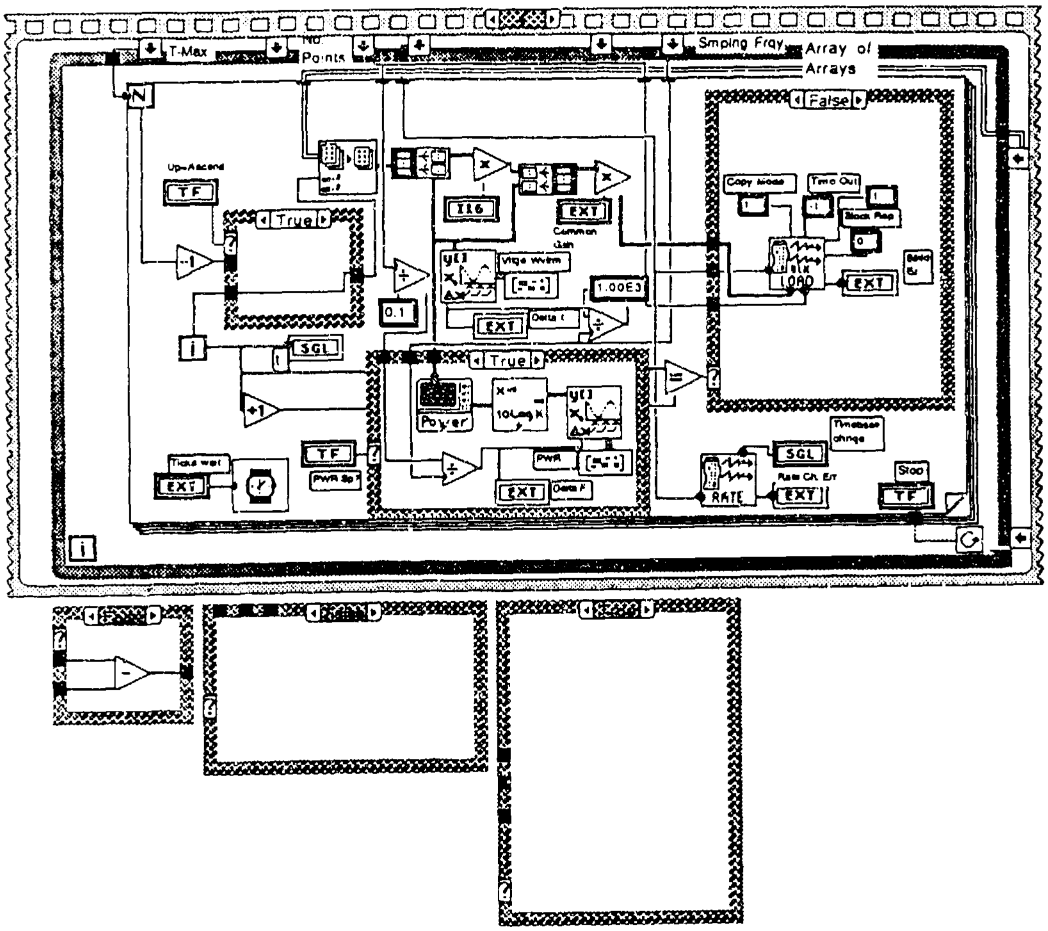


Figure B-5. Frame 4 for Illusory Pitch/Sines. The While Loop inside Frame 4 supports the repeated cycling of the functions contained within it. The direction of pitch change, the interaural intensity difference, loudness of the PC and rate of change in the pitch is controlled within the While Loop. The display of the power spectrum is also controlled.

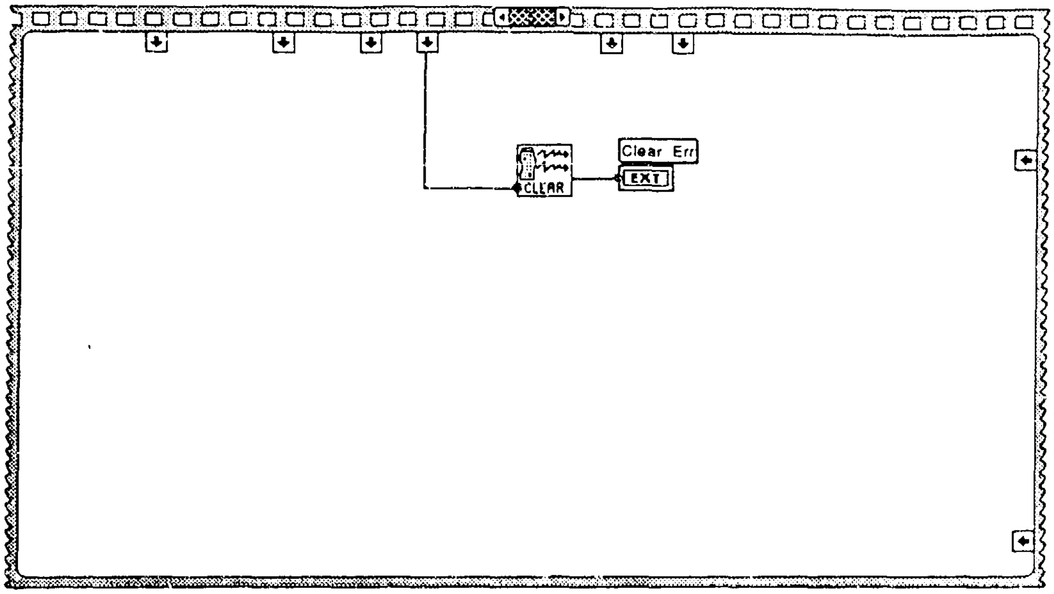


Figure B-6. Frame 5 of Illusory Pitch/Sines. When the stop switch is closed, the program exits the While Loop, moves to Frame 5, executes the sub-VI, CLEAR, and stops.

Illusory Pitch/Noise
Wednesday, June 10, 1992 10:21 AM

Page 1

Front Panel

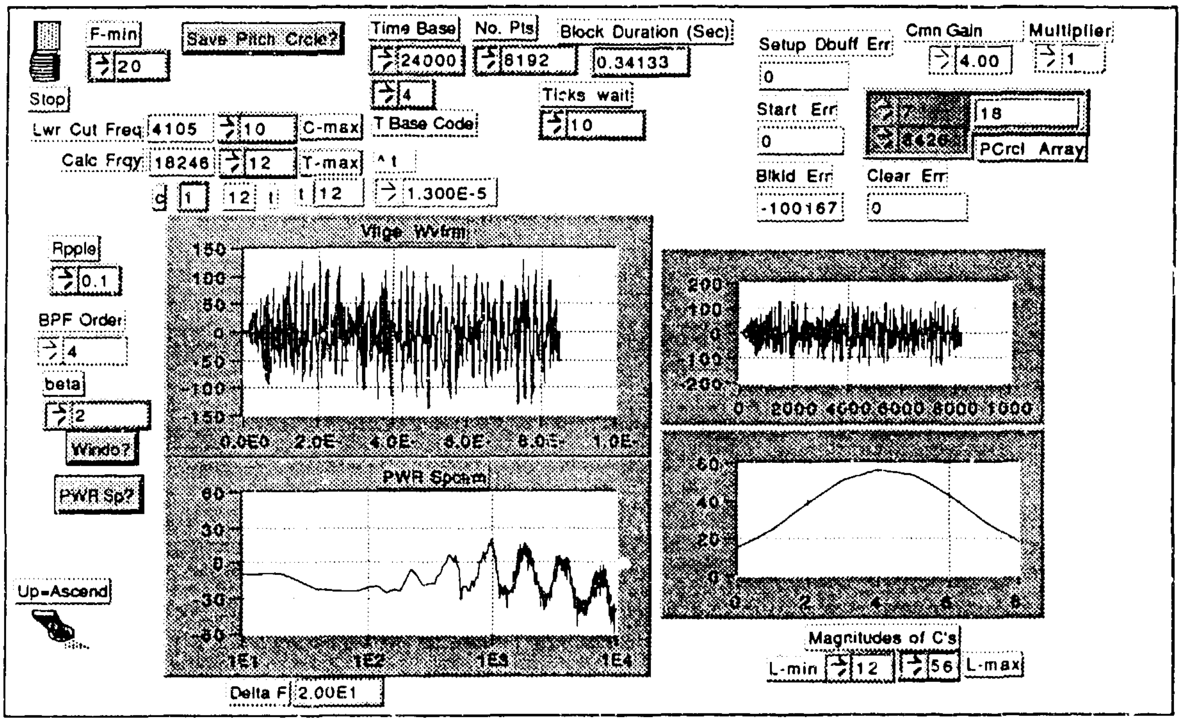


Figure B-7. Front Panel of Illusory Pitch/Noise. Note the contrast of the power spectrum for noise with that shown for sines (Fig. B-1).

Block Diagram

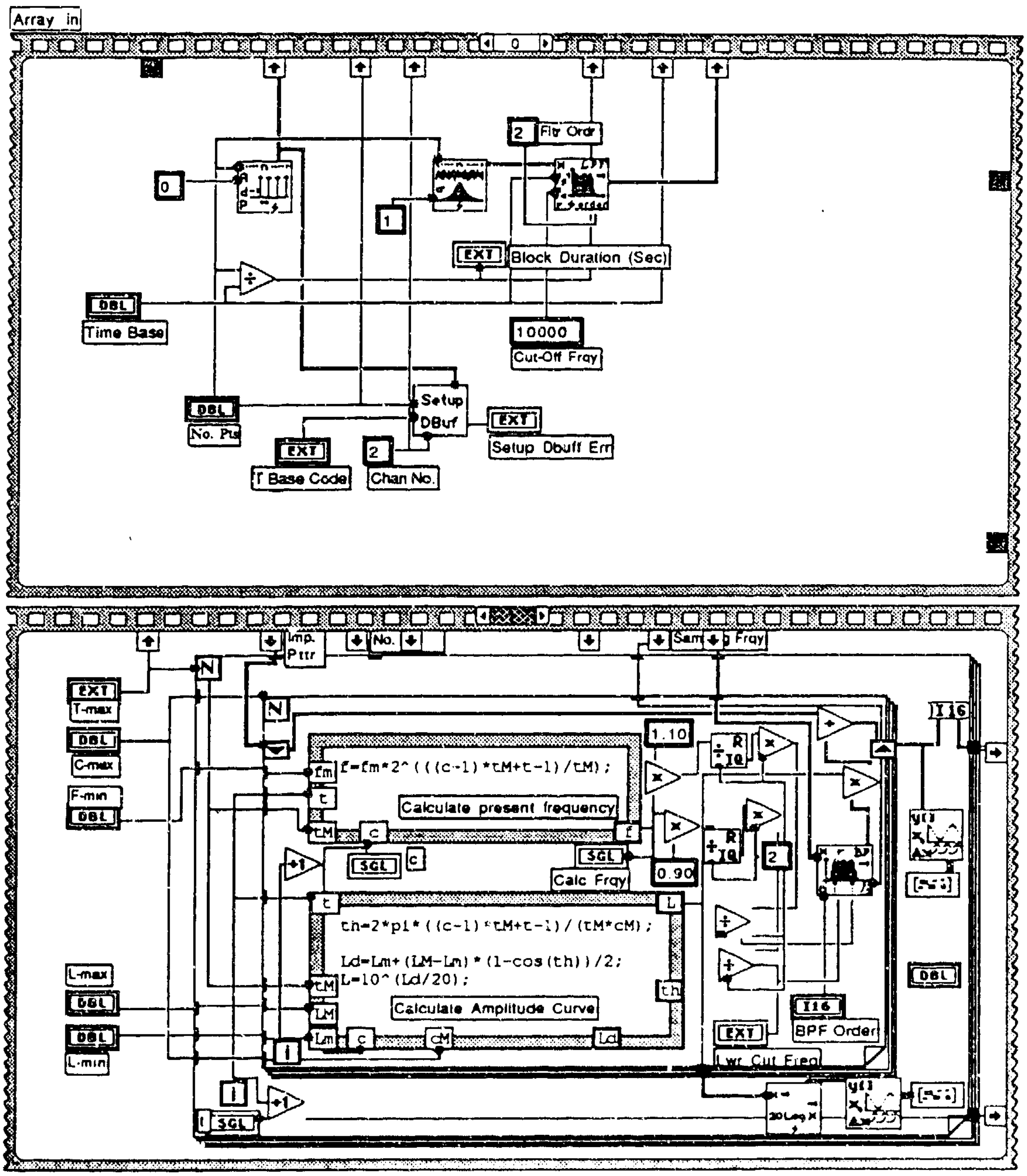


Figure B-8. Frames 0 and 1 for Illusory Pitch/Noise--similar to Figure B-2 except for establishing the limits of band-pass noise in Frame 1. The frequency output from the formula node is multiplied by 0.9 to establish the lower frequency limit and by 1.1 to establish the upper frequency limit of the noise band. See text for development.

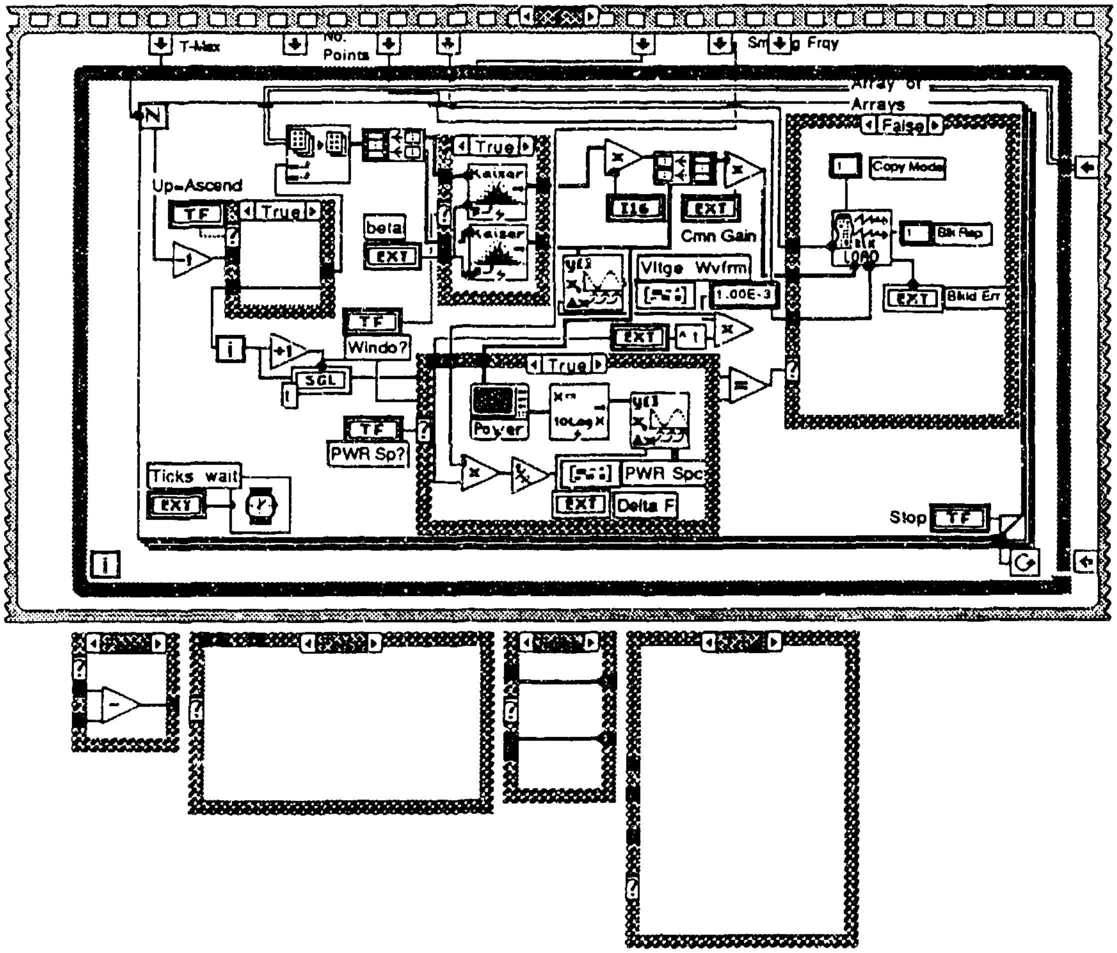


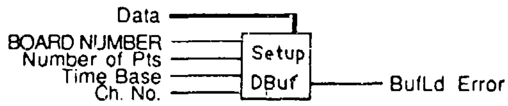
Figure B-9. Frame 4 for Illusory Pitch/Noise--similar to Figure B-5. The window must be included for each channel.

APPENDIX C

LabVIEW Representation of Miscellaneous Virtual Instruments (VIs)

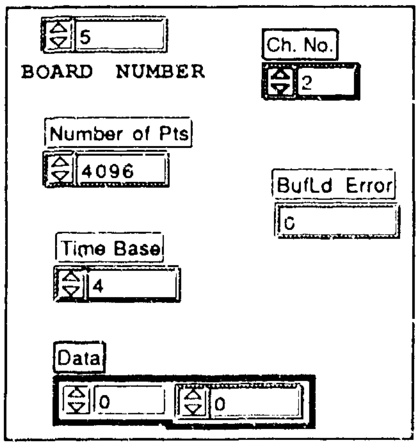
Sub-VIs may be created for various reasons. One of the most useful features of LabVIEW is that its many sub-VIs that can be connected in various ways and used as the elements from which instruments are constructed. Other reasons to create sub-VIs are to save space on the block diagram, or to combine functions that are to be used again and again into one object that can be easily inserted into one's diagram, instead of wiring again each instance of use. The miscellaneous sub-VIs shown here are simply conveniences that seemed useful as the programs were being developed.

Connector Pane



Setup Dble Buffering VI

Front Panel



Block Diagram

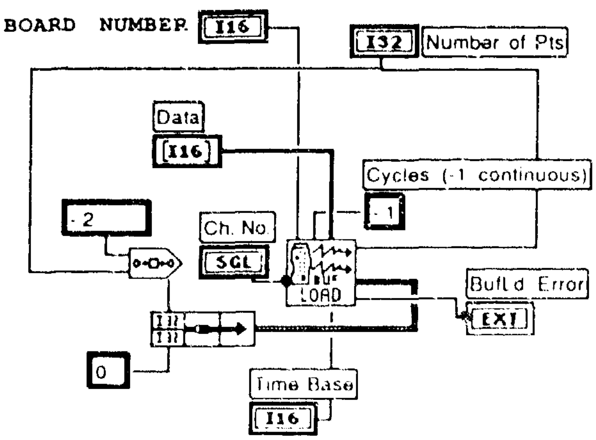
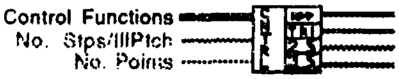


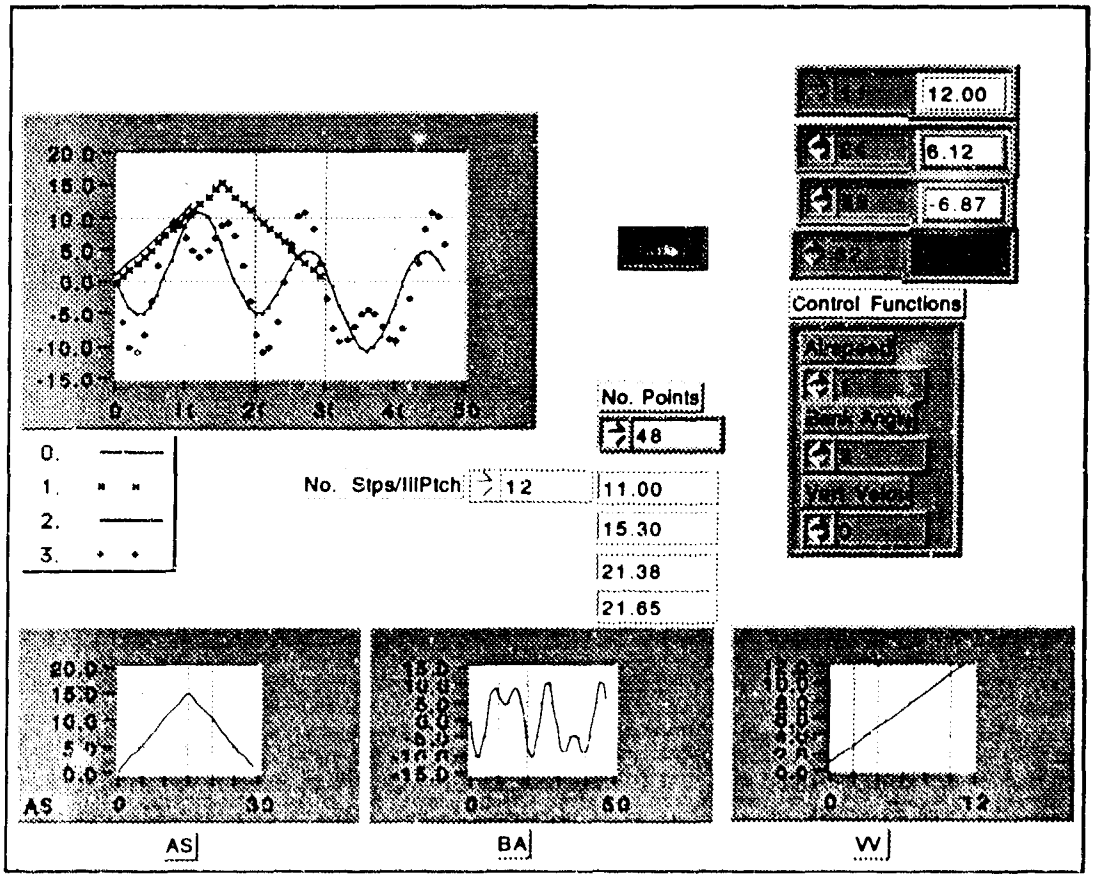
Figure C-1. Connector Pane, Front Panel and Block Diagram for sub-VI, Setup Dble Buffering.

Connector Pane



Pattern Gen

Front Panel



Block Diagram

Figure C-2. Connector Pane and Front Panel for sub-VI, Pattern Gen.

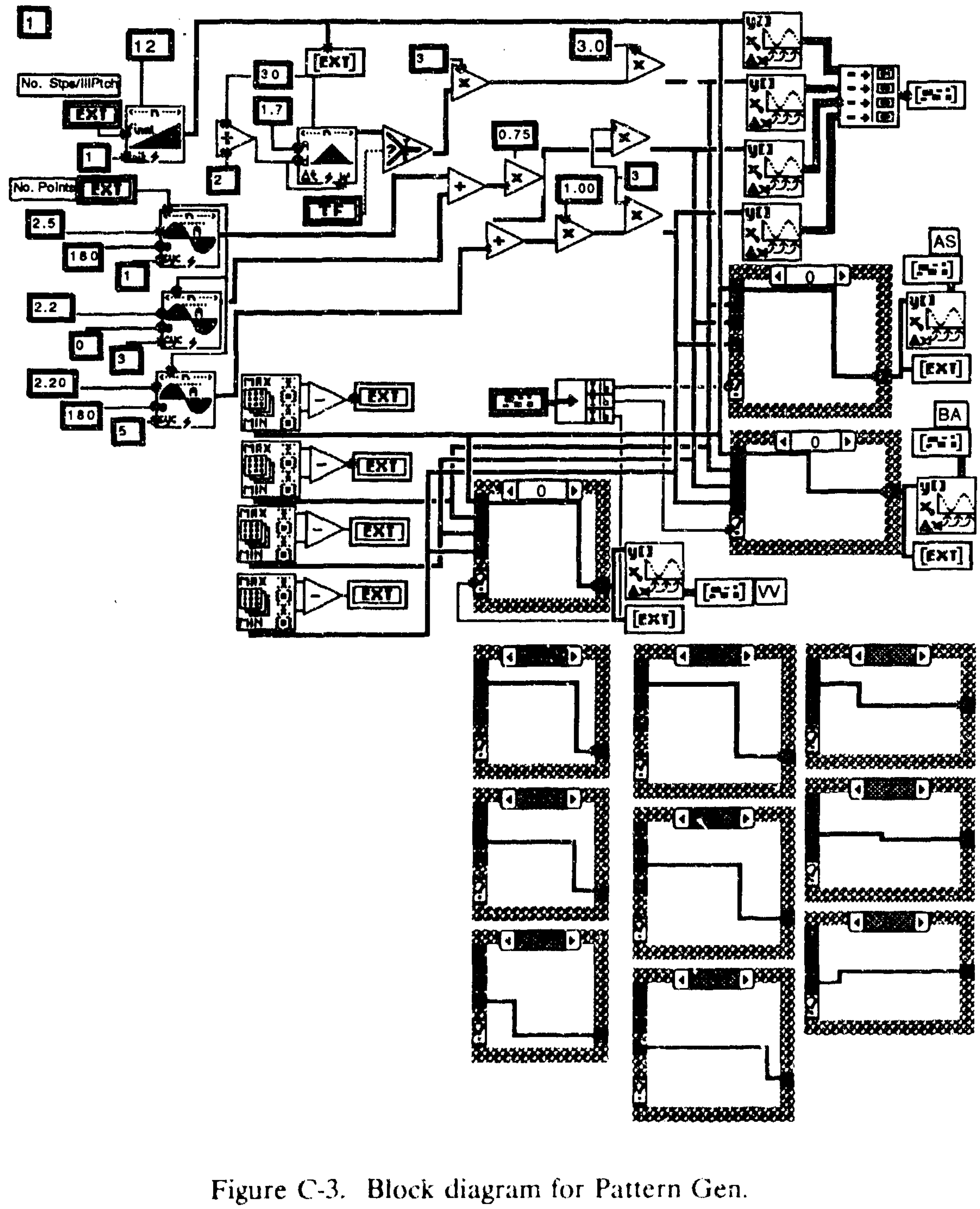
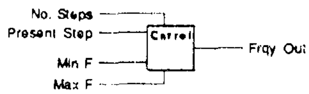


Figure C-3. Block diagram for Pattern Gen.

VI-control(ep)
Thursday, September 24, 1992 3:59 PM

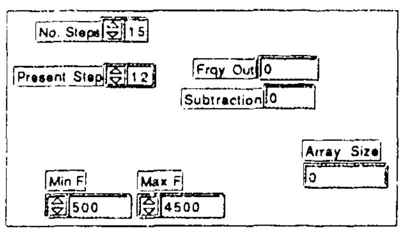
Page 1

Connector Pane



VI-control(ep)

Front Panel



Block Diagram

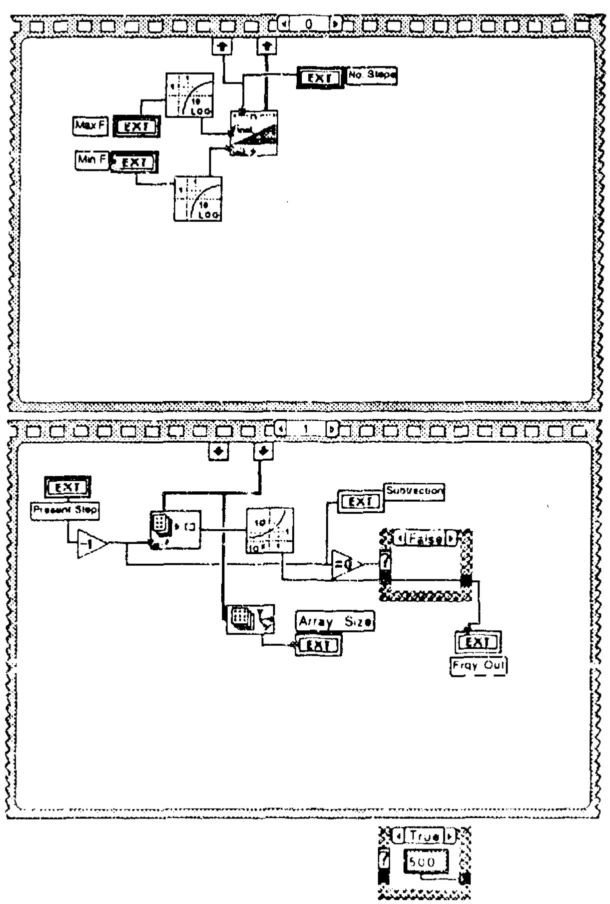
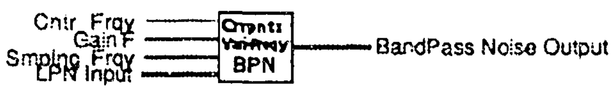


Figure C-4. Connector Pane, Front Panel and Block diagram for sub-VI, control.

COMP BPN
Friday, October 2, 1992 1:30 PM

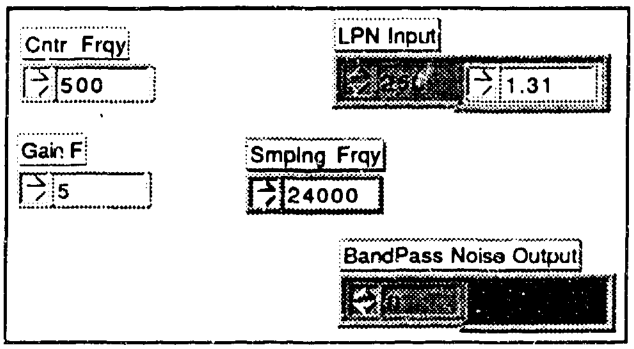
Page 1

Connector Pane



COMP BPN

Front Panel



Block Diagram

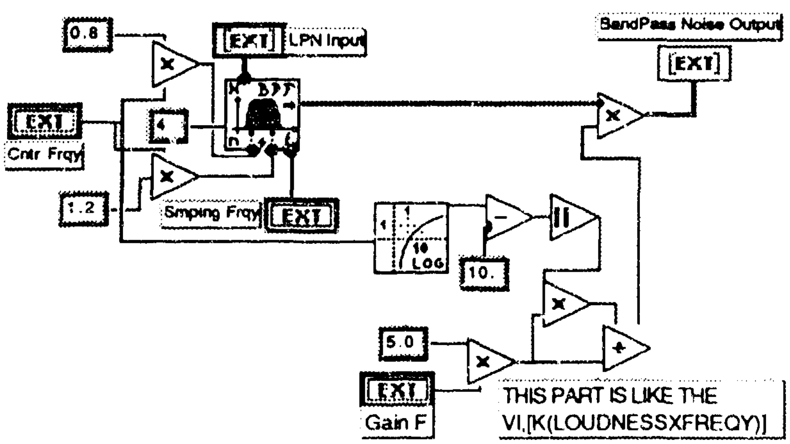
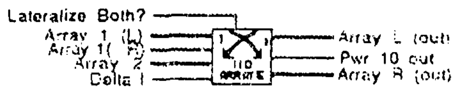


Figure C-5. Connector Pane, Front Panel and block diagram for sub-VI, Comp BPN. This sub-VI is used to generate the center frequencies for the band-pass noise from the values output by Pattern Gen.

SW1/SW2
Friday, November 6, 1992 12:00 PM

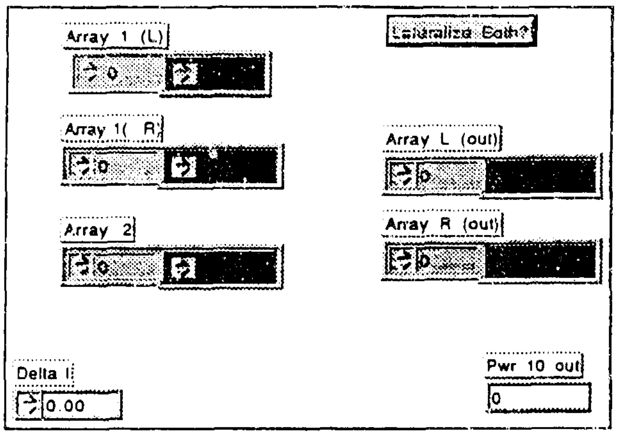
Page 1

Connector Pane



SW1/SW2

Front Panel



Block Diagram

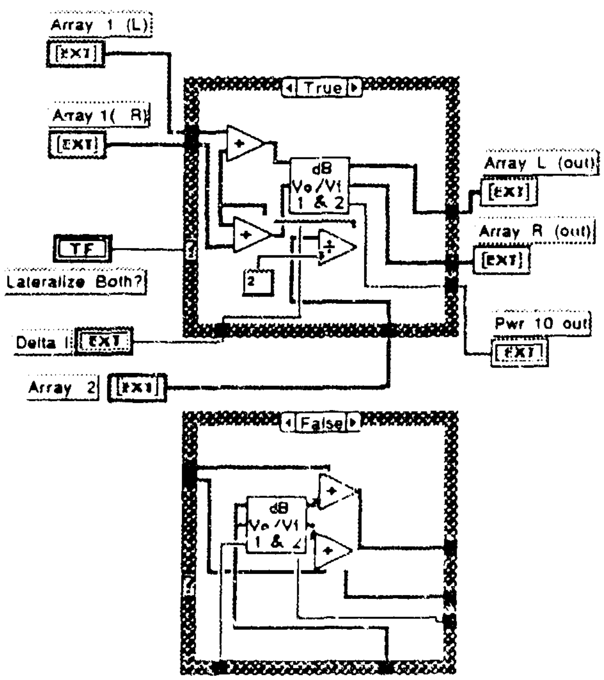
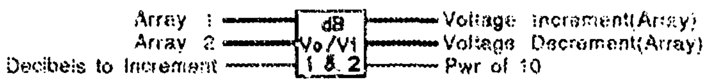


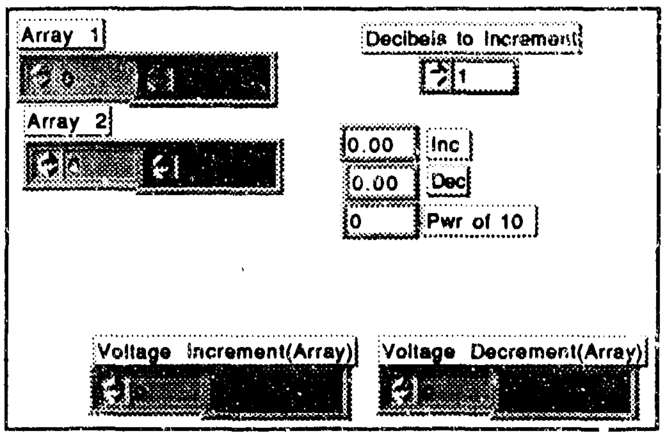
Figure C-6. Connector Pane, Front Panel and block diagram for sub-VI, SW1/SW2. The sub-VI is used to select one or both sets of two arrays each for lateralizing. It is used with the sub-VI, Vo/Vi, to switch a voltage difference between pairs of earphones.

Connector Pane



Vo/Vi Chs 1&2/dB

Front Panel



Block Diagram

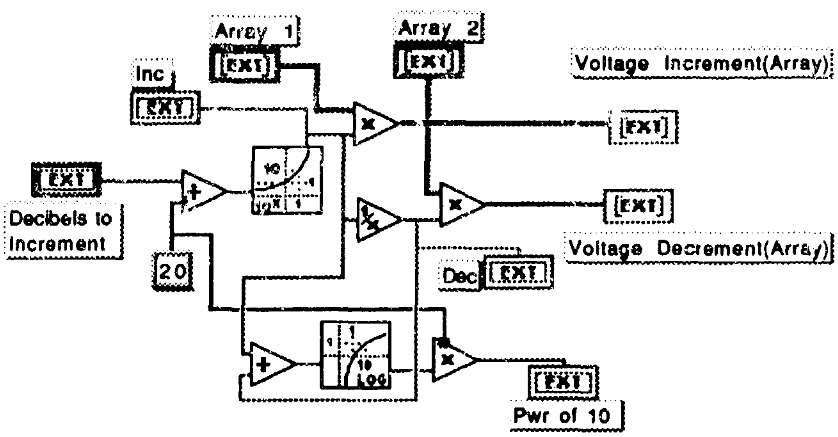
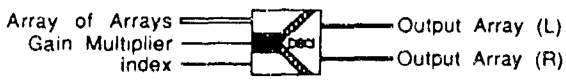


Figure C-7. Connector Pane, Front panel and block diagram for sub-VI, Vo/Vi Chs 1 & 2/dB. The sub-VI is used to convert a number, considered as decibels, to a voltage difference to be delivered to a pair of earphones.

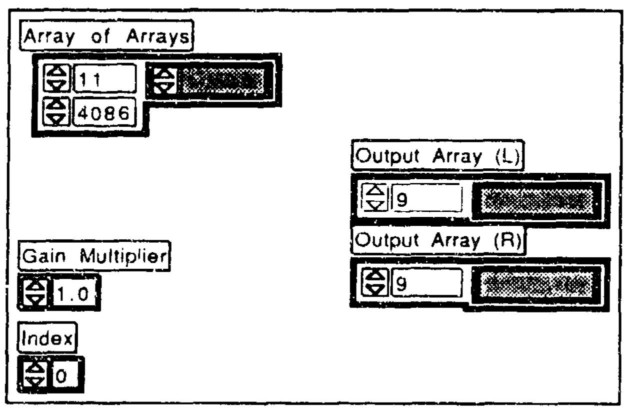
DECIM
Wednesday, July 8, 1992 12:30 PM

Connector Pane



DECIM

Front Panel



Block Diagram

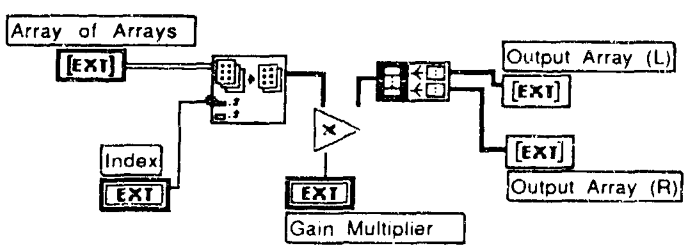


Figure C-8. Connector Pane, Front panel and Block diagram for sub-VI, DECIM. Used to separate an interleaved array into arrays and to add a gain to one channel.